

US-PAT-NO: 5872743

DOCUMENT-IDENTIFIER: US 5872743 A

****See image for Certificate of Correction****

TITLE: Method and apparatus for locating the user of a computer system

----- KWIC -----

Abstract Text - ABTX (1):

A method and apparatus for locating the user of a computer system and adjusting the sound from the system speakers. The computer system includes a computer, two speakers, a microphone, a signal processor, and an audio processor. The speakers emit a sound which is reflected from the user and other objects in the vicinity of the system. The invention includes an algorithm for determining which of the reflected sounds had been reflected from the user of the computer system. The character of the sound emitted by the computer system is then adjusted to enhance the sound effects at the location of the user.

US Patent No. - PN (1):

5872743

Brief Summary Text - BSTX (2):

The present invention relates to methods and apparatus for determining the location of the user of a computer, and in particular to adjusting the sound controlled by the computer to enhance the stereo effect at the location of the user.

Detailed Description Text - DETX (19):

The difference between the first elapsed time and the second elapsed time is thus used to adjust the audio content coloration of sounds being transmitted from the speakers during the execution of software 24, for example during the playing of a game. Audio content coloration parameters of the sound that may be adjusted includes the intensity or amplitude, phase content, signal delays, equalization, or other audio content parameters. The shorter elapsed time indicates that the user is closer to that particular speaker. The intensity, phase, time delay, or other audio parameter, of the sound from that speaker may then be reduced, or alternatively the parameter of the sound from the other speaker may be increased as indicated in block 165. Thus, the present invention can modify the predetermined stereo balance, phase, time delay, or other audio parameter of the sound data within software 24 based upon the location of the user.

Claims Text - CLTX (21):

adjusting the intensity of transmitted sounds based upon the location of the portion of the person.

US-PAT-NO: 5872743

DOCUMENT-IDENTIFIER: US 5872743 A

See image for Certificate of Correction

TITLE: Method and apparatus for locating the user of a computer system

----- KWIC -----

Abstract Text - ABTX (1):

A method and apparatus for locating the user of a computer system and adjusting the sound from the system speakers. The computer system includes a computer, two speakers, a microphone, a signal processor, and an audio processor. The speakers emit a sound which is reflected from the user and other objects in the vicinity of the system. The invention includes an algorithm for determining which of the reflected sounds had been reflected from the user of the computer system. The character of the sound emitted by the computer system is then adjusted to enhance the sound effects at the location of the user.

US Patent No. - PN (1):

5872743

Brief Summary Text - BSTX (2):

The present invention relates to methods and apparatus for determining the location of the user of a computer, and in particular to adjusting the sound controlled by the computer to enhance the stereo effect at the location of the user.

Detailed Description Text - DETX (19):

The difference between the first elapsed time and the second elapsed time is thus used to adjust the audio content coloration of sounds being transmitted from the speakers during the execution of software 24, for example during the playing of a game. Audio content coloration parameters of the sound that may be adjusted includes the intensity or amplitude, phase content, signal delays, equalization, or other audio content parameters. The shorter elapsed time indicates that the user is closer to that particular speaker. The intensity, phase, time delay, or other audio parameter, of the sound from that speaker may then be reduced, or alternatively the parameter of the sound from the other



US05872743A

United States Patent (19)

Patent Number: 5,872,743

Maxwell

Date of Patent: Feb. 16, 1999

(54) METHOD AND APPARATUS FOR LOCATING THE USER OF A COMPUTER SYSTEM

(75) Inventor: Conrad A. Maxwell, Irvine, Calif.

(73) Assignee: VLSI Technology, Inc., San Jose, Calif.

(21) Appl. No.: 31,080

(22) Filed: Feb. 10, 1998

(51) Int. Cl.⁷ G01S 18/00

(52) U.S. Cl. 367/96; 367/99

(53) Field of Search 367/96, 99; 381/103

(56) References Cited

U.S. PATENT DOCUMENTS

4,556,354 5/1981 Hagan et al. 367/101
4,614,532 3/1988 Ruff et al. 179/1.8
4,823,301 4/1988 Schwartz 381/103
5,107,748 4/1992 Bauer 364/20
5,193,197 11/1993 Griesinger 361/25
5,214,615 5/1993 Bauer 367/128
5,233,644 8/1993 Yanggren et al. 381/89

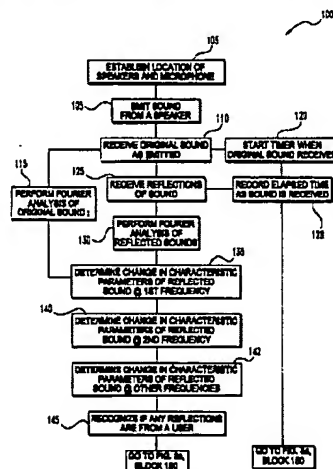
5,255,326 10/1993 Saravanan 381/110
5,386,478 1/1995 Plachet 381/103
5,412,619 5/1995 Bauer 367/128
5,467,401 11/1995 Nagashima et al. 381/83
5,544,349 8/1998 Opta 381/83
5,588,083 12/1998 Edgar 381/24

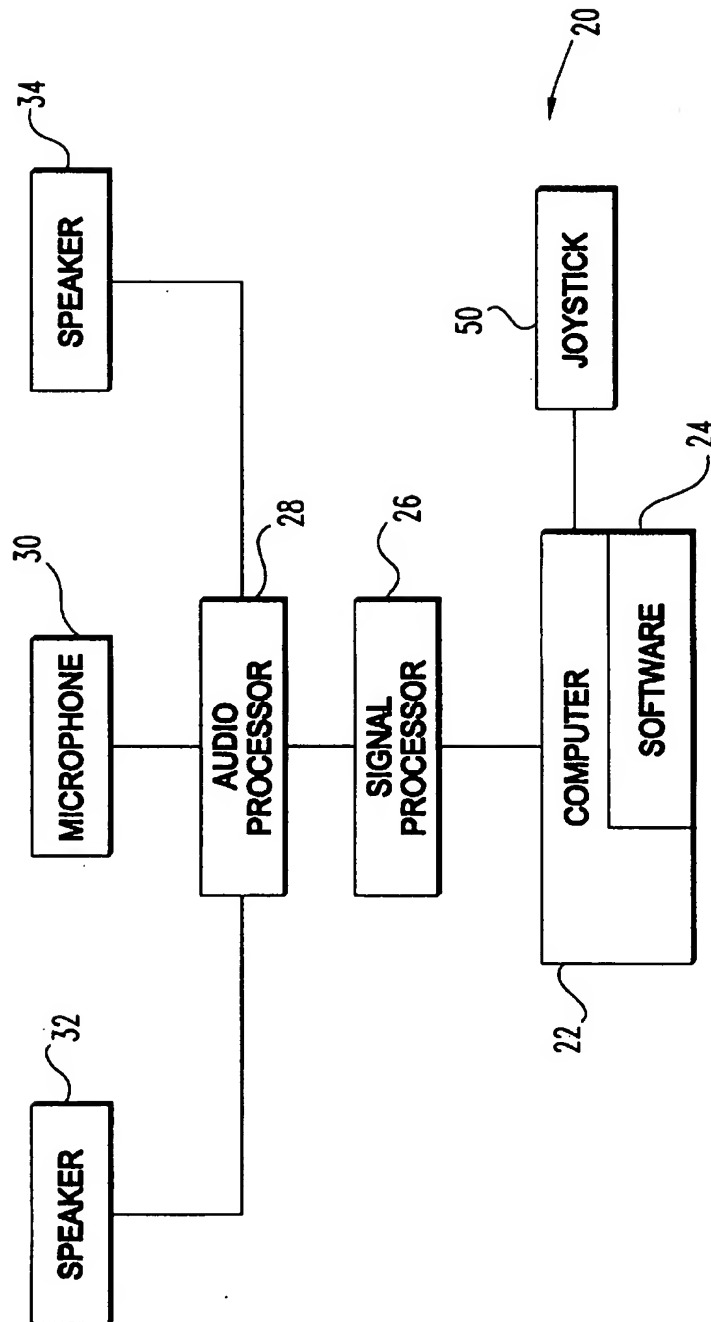
Primary Examiner—Daniel T. Plimic
Attorney Agent, or Firm—Woodward, Emhardt, Naughton,
Modary & McKee

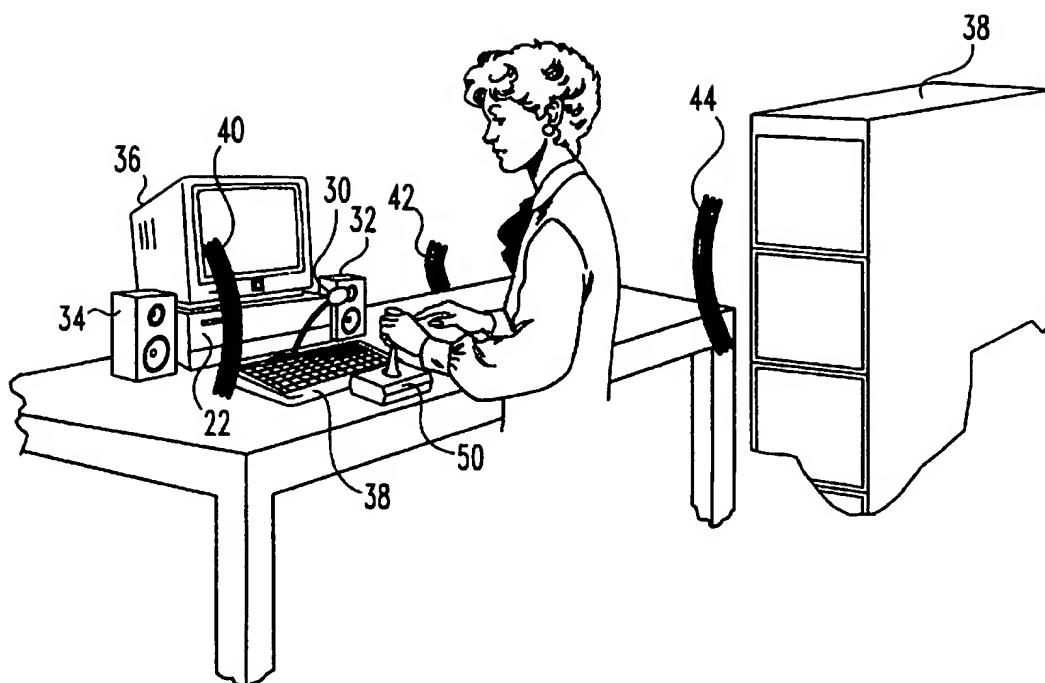
ABSTRACT

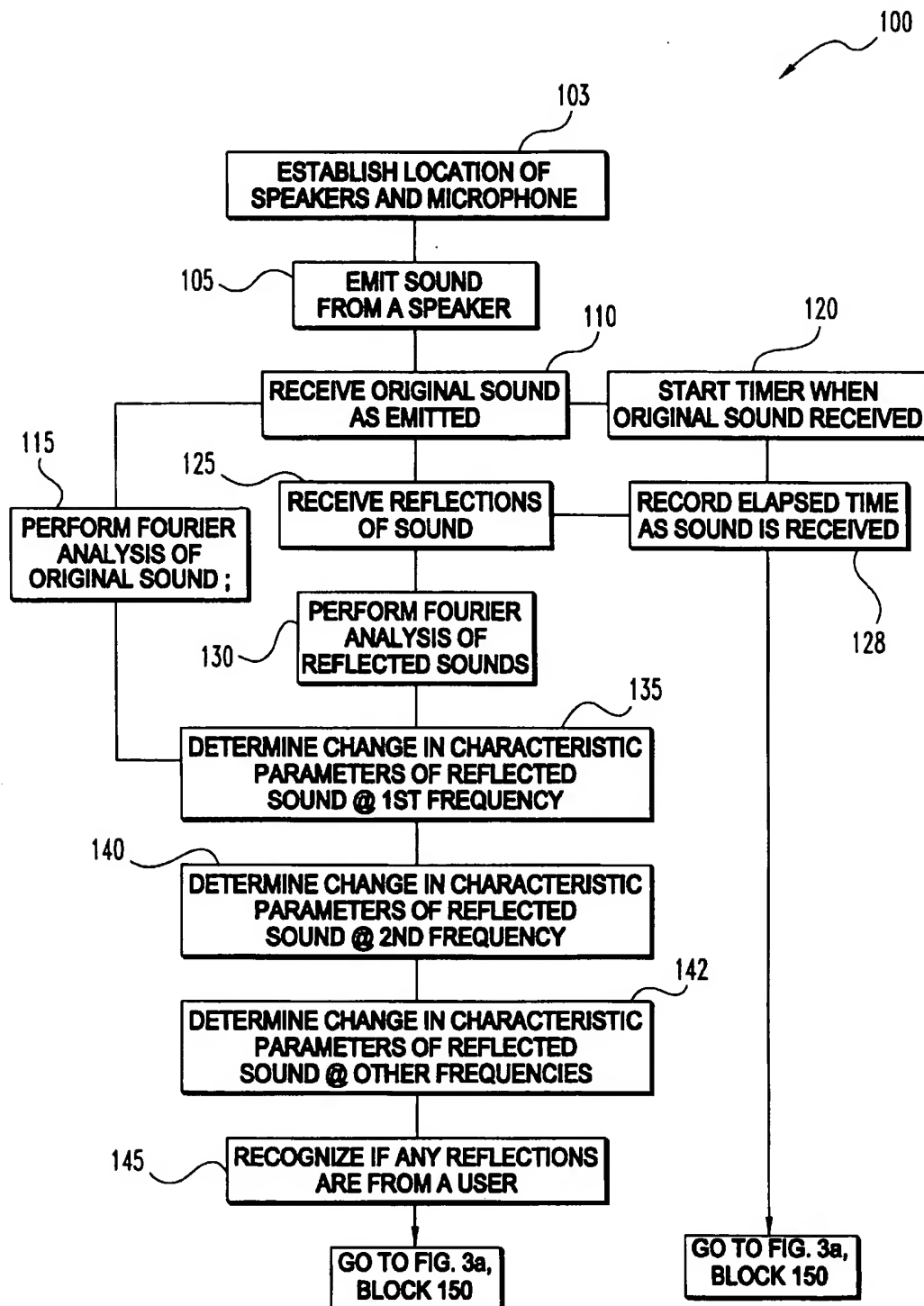
A method and apparatus for locating the user of a computer system and adjusting the sound from the system speakers. The computer system includes a computer, two speakers, a microphone, a signal processor, and an audio processor. The speakers emit a sound which is reflected from the user and other objects in the vicinity of the system. The invention includes an algorithm for determining which of the reflected sounds had been reflected from the user of the computer system. The character of the sound emitted by the computer system is then adjusted to enhance the sound effects at the location of the user.

13 Claims, 4 Drawing Sheets



**Fig. 1**

**Fig. 2**

**Fig. 3a**

US-PAT-NO: 5872743

DOCUMENT-IDENTIFIER: US 5872743 A

****See image for Certificate of Correction****

TITLE: Method and apparatus for locating the user of a computer system

----- KWIC -----

US Patent No. - PN (1):

5872743

Application Filing Date - AD (1):

19980210

Detailed Description Text - DETX (11):

The sound emitted from speaker 32 or 34 varies in amplitude and phase from the input electrical signal sent by audio processor 28. For this reason the original sound as emitted from the speaker is received by microphone 30 as indicated in block 110. This original sound is received by microphone 30 which generates a returned electrical signal to processor 28. This returned electrical signal is then passed on as either an analog signal or a digital signal to signal processor 26, which performs a Fourier analysis of the original sound as indicated by block 115. The Fourier analysis is useful for calculating a representative of the returned electrical signal as a plurality of discrete frequency components, each with a characteristic amplitude. The present invention thus establishes a baseline Fourier analysis of the original sound as emitted by the speaker. This baseline analysis accounts for transmission effects within speakers 32 and 34 and reception effects within microphone 30. The reception of the original signal by microphone 30 also initializes and starts a timer as indicated by block 120.

Detailed Description Text - DETX (14):

Various characteristic parameters such as amplitude, phase shift, or time delay are altered and modified by various objects based upon the characteristics of the surface from which the sound is reflected. Hard surfaces such as those on cabinet 38 will reflect a significant portion of the sound pressure wave incident thereto. This is especially true at higher frequencies, for example frequencies in excess of 5 KHZ, and especially those frequencies in the ultrasonic range. In contrast, the rounded shape and generally soft surface of a human user of system 20 reflects a sound pressure wave 42 that is more attenuated than sound wave 44. For example, emitted sounds in excess of 5 KHZ, and especially those sounds at ultrasonic frequencies, are significantly attenuated after reflection from the human user. A Fourier analysis is performed for each reflected sound received by microphone 30.

Detailed Description Text - DETX (19):

The difference between the first elapsed time and the second elapsed time is

thus used to adjust the audio content coloration of sounds being transmitted from the speakers during the execution of software 24, for example during the playing of a game. Audio content coloration parameters of the sound that may be adjusted includes the intensity or amplitude, phase content, signal delays, equalization, or other audio content parameters. The shorter elapsed time indicates that the user is closer to that particular speaker. The intensity, phase, time delay, or other audio parameter, of the sound from that speaker may then be reduced, or alternatively the parameter of the sound from the other speaker may be increased as indicated in block 165. Thus, the present invention can modify the predetermined stereo balance, phase, time delay, or other audio parameter of the sound data within software 24 based upon the location of the user.

US-PAT-NO: 5872743

DOCUMENT-IDENTIFIER: US 5872743 A

****See image for Certificate of Correction****

TITLE: Method and apparatus for locating the user of a computer system

----- KWIC -----

US Patent No. - PN (1):

5872743

Brief Summary Text - BSTX (8):

One aspect of the present invention concerns a method of locating a person using a computer. The method includes emitting a first sound comprising a plurality of frequencies, and receiving a first reflected sound in response to the first sound being reflected from an object. The amplitude of the emitted first sound is calculated at a plurality of frequencies, and the amplitude of the reflected first sound is calculated at a plurality of frequencies. The amplitude of the emitted first sound is compared to the amplitude of the reflected first sound for at least two of the frequencies. The method includes recognizing from the comparison if the object is a portion of a person.

Detailed Description Text - DETX (4):

Computer system 20 also includes a signal processor 26 receiving commands from and providing data to computer 22. Signal processor 26 may be a separate circuit card within computer 22, a portion of a circuit card within computer 22, a chip within computer 22, or embodied within a chip such as the central processor unit that performs other functions within computer 22. Signal processor 26 is capable of constructing electrical signals to be provided to audio processor 28, and is also capable of receiving electrical signals from audio processor 28. Signal processor 26 may transmit and receive this information from audio processor 28 in either analog or digital fashion. Signal processor 26 is capable of constructing signals or analyzing signals from about 50 HZ to about 25 KHZ. Signal processor 26 includes the hardware and software necessary to perform Fourier analysis within this frequency range. Processor 26 is preferably housed within computer 22, but may also be a stand alone module. Various functions described herein for processor 26 may also be performed within the central processing unit. It is preferable that processor 26 include at least one equalizer for modifying the frequency content of sound data received from software 24 within computer 22. For example, this equalizer can boost or reduce the amplitude of a particular frequency band.

Detailed Description Text - DETX (11):

The sound emitted from speaker 32 or 34 varies in amplitude and phase from the input electrical signal sent by audio processor 28. For this reason the original sound as emitted from the speaker is received by microphone 30 as indicated in block 110. This original sound is received by microphone 30 which generates a returned electrical signal to processor 28. This returned electrical signal is then passed on as either an analog signal or a digital

signal to signal processor 26, which performs a Fourier analysis of the original sound as indicated by block 115. The Fourier analysis is useful for calculating a representative of the returned electrical signal as a plurality of discrete frequency components, each with a characteristic amplitude. The present invention thus establishes a baseline Fourier analysis of the original sound as emitted by the speaker. This baseline analysis accounts for transmission effects within speakers 32 and 34 and reception effects within microphone 30. The reception of the original signal by microphone 30 also initializes and starts a timer as indicated by block 120.

Detailed Description Text - DETX (13):

Microphone 30 generates a returned electrical signal comparable to reflected sound 44. This returned electrical signal is received by processor 28, which then passes this data to signal processor 26 wherein Fourier analysis of the reflected sound is performed in order to calculate the characteristic parameters of the sound at the various frequency components (block 130). In general, there is a reduction in amplitude at all frequencies because of the general expansion of the sound pressure waves as they travel the distance from a speaker to a reflecting object and back to microphone 30.

Detailed Description Text - DETX (14):

Various characteristic parameters such as amplitude, phase shift, or time delay are altered and modified by various objects based upon the characteristics of the surface from which the sound is reflected. Hard surfaces such as those on cabinet 38 will reflect a significant portion of the sound pressure wave incident thereto. This is especially true at higher frequencies, for example frequencies in excess of 5 KHZ, and especially those frequencies in the ultrasonic range. In contrast, the rounded shape and generally soft surface of a human user of system 20 reflects a sound pressure wave 42 that is more attenuated than sound wave 44. For example, emitted sounds in excess of 5 KHZ, and especially those sounds at ultrasonic frequencies, are significantly attenuated after reflection from the human user. A Fourier analysis is performed for each reflected sound received by microphone 30.

Detailed Description Text - DETX (15):

Block 135 indicates comparison of the amplitudes of the frequency components from the Fourier analysis of the original sound to the amplitudes of the same frequency components from the Fourier analysis of the reflected sounds, for example at a first frequency component below 1 KHZ. The reflected sound is generally lower in amplitude at the particular frequency component than the originally emitted sound. Subtracting the amplitude of the reflected sound from the amplitude of the emitted sound quantifies a first amplitude reduction associated with the particular reflected sound at this first low frequency. Block 140 indicates a similar comparison of the amplitudes of the frequency components from the Fourier analysis of the original sound to the Fourier analysis of the reflected sound at a second frequency, for example at a frequency above about 5 KHZ. Subtraction of the amplitude of the reflected sound from the amplitude of the emitted sound quantifies a second amplitude reduction at this second high frequency.

Detailed Description Text - DETX (16):

The characteristic parameters such as amplitude, phase shift, or time delay of each reflected sound 42 and 44, are compared to the characteristic

parameters of the originally emitted sound 40 as indicated in Blocks 135, 140, and 142. In general, reflected sounds have a reduction in the characteristic parameter of amplitude at each frequency component as compared to that same frequency component in the originally emitted sound 40. However, the reduction in amplitude of reflected sounds from a human user is greater at higher frequencies than the reduction in amplitude of reflected sounds from objects such as cabinets. On the other hand, lower frequency sounds are also reduced after reflection from a human user, but not as much as the amplitude reduction at higher frequencies. Thus, reflected sound 44 has a reduction in amplitude across frequencies that is more uniform than the reduction in amplitude in sound 42. In contrast, reflected sound 42 has more reduction at higher frequencies than at lower frequencies. Thus, the first amplitude reduction associated with sound wave 44 is of a magnitude similar to the second amplitude reduction associated with sound wave 44. In contrast, the second amplitude reduction associated with the sound wave 42 is of a magnitude greater than the first amplitude reduction associated with sound wave 42. A second amplitude reduction calculated to be at least 1 decibel less than the first amplitude reduction is sufficient for algorithm 100 to recognize sound wave 42 as being reflected from a human user (block 145). In other embodiments of the present invention phase shifting and time delays that correspond to the characteristics of the reflection surface are used to discriminate the human user from other objects.

Detailed Description Text - DETX (17):

After a particular reflected sound wave is recognized from the comparison of the amplitude reductions as being reflected from a portion of a person, then as indicated in block 150 the elapsed time for the particular reflected sound wave is noted. The aforementioned logic of blocks 105 through 150 is repeated for the other speaker, as noted in block 155. The present invention also contemplates those embodiments in which sounds are emitted simultaneously from speakers 32 and 34 during block 105. In those embodiments the frequencies emitted by speaker 32 are different from the frequencies emitted by speaker 34. By emitting different frequencies from the two speakers 32 and 34 it is possible to implement logic blocks 105-150 in nearly simultaneous fashion. The use of different frequencies permits sounds to be simultaneously emitted from speakers 32 and 34 and simultaneously received by microphone 30 without processor 28 or processor 26 encountering ambiguity about the source of the emitted sound.

Detailed Description Text - DETX (18):

After again comparing the amplitude reductions of the emitted sounds to the amplitude reductions of the reflected sounds and recognizing from the comparison if an object is a portion of a person, another elapsed time is noted for the second speaker. Thus there is a first elapsed time associated with the distance from the user to the first speaker, and a second elapsed time associated with the distance from the user to the second speaker. The location of the user can then be determined from knowledge of the first and second elapsed times. Knowing the speed of sound within a typical room environment and also knowing the relative geometric locations of the speakers 32 and 34 and microphone 30, it is possible to redefine each elapsed time as a radial distance from the respective speaker in a method known to those of ordinary skill in the art as triangulation. The intersection of the radial distances from each speaker is the location of the user (block 160).

Detailed Description Text - DETX (19):

The difference between the first elapsed time and the second elapsed time is thus used to adjust the audio content coloration of sounds being transmitted from the speakers during the execution of software 24, for example during the playing of a game. Audio content coloration parameters of the sound that may be **adjusted includes the intensity or amplitude**, phase content, signal delays, equalization, or other audio content parameters. The shorter elapsed time indicates that the user is closer to that particular speaker. The intensity, phase, time delay, or other audio parameter, of the sound from that speaker may then be reduced, or alternatively the parameter of the sound from the other speaker may be increased as indicated in block 165. Thus, the present invention can modify the predetermined stereo balance, phase, time delay, or other audio parameter of the sound data within software 24 based upon the location of the user.

Claims Text - CLTX (3):

calculating the **amplitude** of the first emitted sound at a plurality of frequencies;

Claims Text - CLTX (4):

calculating the **amplitude** of the first reflected sound at a plurality of frequencies;

Claims Text - CLTX (5):

comparing the **amplitude** of the first emitted sound to the **amplitude** of the first reflected sound for at least two of the frequencies; and

Claims Text - CLTX (11):

calculating the **amplitude** of the second emitted sound at a plurality of frequencies;

Claims Text - CLTX (12):

calculating the **amplitude** of the second reflected sound at a plurality of frequencies;

Claims Text - CLTX (13):

comparing the **amplitude** of the second emitted sound to the **amplitude** of the second reflected sound for at least two of the frequencies; and

Claims Text - CLTX (23):

determining a first **amplitude** reduction of the first reflected sound relative to the first emitted sound at a first frequency below about 1 KHZ;

Claims Text - CLTX (24):

determining a second **amplitude** reduction of the first reflected sound relative to the first emitted sound at a second frequency above about 5 KHZ; and

Claims Text - CLTX (25):

calculating the second **amplitude** reduction to be at least 1 decibel less

than the first amplitude reduction.

Claims Text - CLTX (32):

a signal processor capable of determining the amplitude of the input electrical signals at a plurality of frequencies; and

Claims Text - CLTX (39):

determining a first amplitude reduction of the first reflected sound relative to the first emitted sound at a first frequency below about 1 KHZ;

Claims Text - CLTX (40):

determining a second amplitude reduction of the first reflected sound relative to the first emitted sound at a second frequency above about 5 KHZ; and

Claims Text - CLTX (41):

calculating the second amplitude reduction to be at least 1 decibel less than the first amplitude reduction.